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Development and Documentation of the Noise Simulation Facility at DRDC Toronto

Mise au point de l'installation de simulation de bruit à l'agence RDDC de Toronto et documentation

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Abstract

This document covers various topics relating to acoustic research performed by the Noise and Communication Group at DRDC Toronto. It includes a description of both hardware and software developed and used for research studies conducted in support of Canadian Forces operations. The main components of this document are related to the noise simulation facility and its accompanying software suite, including the Modified Rhyme Test and Maximum Likelihood Procedure. The Microsoft Access Noise Database is also discussed.

Résumé

Le présent document traite de divers sujets sur les recherches acoustiques menées par le groupe d'acoustique et des communications de RDDC à Toronto. Il comprend une description du matériel et des logiciels conçus et utilisés dans le cadre des recherches techniques effectuées afin d'appuyer les opérations des Forces canadiennes. Les principaux éléments de ce document portent sur l'installation de simulation de bruit et sur la suite logicielle correspondante, notamment la méthodologie de test Modified Rhyme Test et la procédure du maximum de vraisemblance. La base de données Microsoft Access sur le bruit fait également l'objet de discussions.

Executive Summary

There are many different systems, both hardware and software, used by the Noise and Communications Group at DRDC Toronto. Most of these systems have been designed for the exclusive use of the Comms Group and often have little or no accompanying documentation. This is the fundamental reason for this document: to have written descriptions of the systems used and how they function.

The bulk of the documentation pertains to the noise simulation facility and its control system. The documentation includes a complete description of the current system, how it is used and what its characteristics are. It will be invaluable in the future as work continues in the noise simulation facility.

Also documented here is the Modified Rhyme Test (MRT) and Maximum Likelihood Procedure (MLP). This is a software system used for various experiments. It is used in combination with the noise simulation facility.

The final software project discussed is the PC version (Microsoft Access) of the noise database. The existing Macintosh database was examined and ported to the PC platform.

Sommaire analytique

Le groupe d'acoustique et des communications de RDDC à Toronto utilise un grand nombre de systèmes matériels et logiciels différents. La plupart de ces systèmes ont été conçus pour être utilisés exclusivement par le groupe de communications et comportent peu ou pas de documentation. La raison première de ce document est de décrire les systèmes utilisés et la façon dont ils fonctionnent.

La majorité de la documentation porte sur l'installation de simulation de bruit et sur son système de contrôle. La documentation donne une description complète du système courant, explique la façon dont il est utilisé et ses caractéristiques. Elle aura désormais une très grande valeur parce que le travail continue au niveau de l'installation de simulation de bruit.

Le présent document traite également de Modified Rhyme Test (MRT) et de la procédure du maximum de vraisemblance (MLP). Il s'agit d'un logiciel utilisé de concert avec l'installation de simulation de bruit, de même que pour faire diverses expériences.

Le dernier projet logiciel qui a fait l'objet de discussions est la version PC (Microsoft Access) de la base de données sur le bruit. La base de données Macintosh courante a été examinée et portée sur PC.

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Introduction

This manual contains all the documentation pertaining to the various areas of work done by Trellis Consulting while working on Contract #W7711-017769/001/TOR. The contract covers the period from April 1, 2002 to March 31, 2004.

Within this manual are different sections, each containing information pertaining to a particular area under the contract. Each of those sections is a self-contained document. This main document merely exists to tie all of the smaller documents together in one place.

Section A contains a detailed description of the noise simulation facility sound system.

Section B contains information about the sounds available in the noise simulation facility.

Section C describes the MRT/MLP software developed for use with various protocols for experiments.

Section D outlines the backups in place for the Hearing Research lab system.

Section E is a discussion of how to calibrate direct inputs on the B&K2133.

Section F gives a thorough description of the Microsoft Access database developed to help estimate noise levels under headsets, in various noise environments.

Section G contains portions of the manuals from the equipment used in the noise simulation facility.

Appendix A

The DRDC Toronto Noise Simulation Facility

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1 Introduction

This document describes the system used at DRDC Toronto to power the Noise Simulation Facility. It includes all of the knowledge the author (Garry Dunn) has gained in using the system.

It is by no means an exhaustive description of all the various components of the system. For that, the reader needs to refer to the various manuals for each of the pieces of equipment found in the system (the manuals are contained in the appendices).

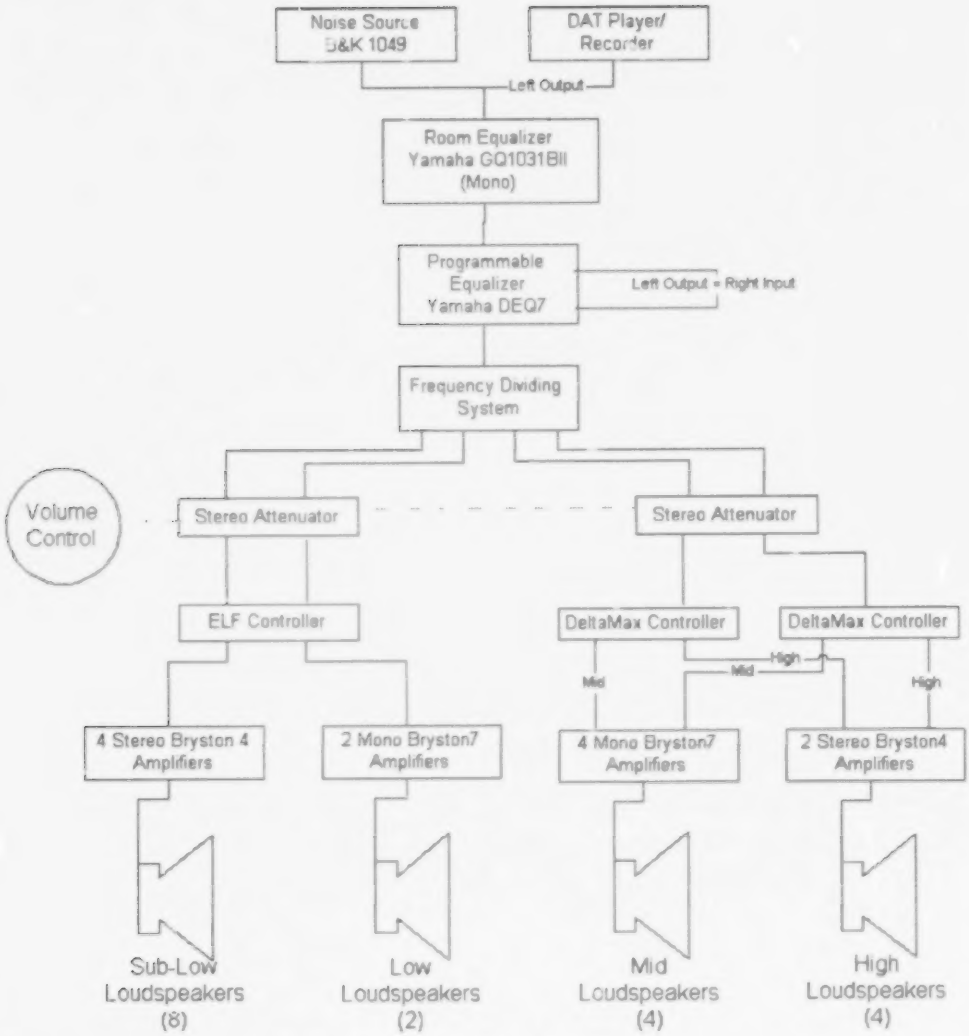
This documentation should suffice for anyone wishing to use the system in its normal day to day operation. How to configure the various noises, as well as how we 'tune' the room to new noise sources are covered. These are, by far the most common things done with the system.

The room itself is 10.55m long, by 6.10m wide, with a height of 3.05m. The reverb characteristics are shown in Section 21 of this manual.

2 System Structure

The diagram below shows how the input signal is processed through the noise system and ultimately delivered to the amplifiers. A brief discussion of each component of the system is included here, however, for complete details, consult the manuals for each device.

An important note is that the entire system deals with a monophonic signal. The signal is equalized and split into 4 frequency bands (high, mid, low and sub-low). Each of those frequency bands is sent to amplifiers to directly drive the speakers.



Starting at the beginning of the entire chain is the input source. This can either be the B&K Noise Generator or the DAT player. From there, the signal is processed by a graphic equalizer. The settings on the graphic equalizer have never moved (to my knowledge). I don't know how they were arrived at.

From the graphic EQ, the signal is routed to the DEQ7. The DEQ7 is a 2 channel device and both channels are used at this stage. The input to channel 1 is used as the primary input. The output of channel 1 is routed to the input of channel 2. The output of channel 2 is the signal which continues. This is an important note since it means the settings on the DEQ7 have twice the effect you might think (due to the serialization). This means that if the DEQ7 is programmed to have 6 dB of attenuation at a particular frequency, it is actually performing 12dB of attenuation (6dB on the left channel and 6dB on the right channel).

After the DEQ7, the signal goes to the BSS frequency dividing system. This component does the fundamental splitting into the 4 frequency bands. After leaving this component, four independent

signals are present (together they form the entire frequency band). Each signal goes to a digitally controlled attenuator (DCA). This is the location in the chain where the operator has control over the level in the room (via the Oxmoor attenuators).

The outputs of the DCA's go to different locations, depending on the frequency band. The highs and mids are routed to the Delta Max controllers for the Electro Voice speakers. That system has its own frequency dividing system which gives outputs for the small and large drivers. The high outputs drive both the left and right side of a Bryston 4, running in stereo mode. From two Bryston 4's, four outputs are obtained, one for each of the 4 high frequency drivers on the Electro Voice speakers. The low outputs drive four separate Bryston 7 amplifiers running in mono mode, with each amplifier driving one driver each.

The other 2 outputs from the DCA's go to the ELF component. This component is a two channel system which further splits the low frequency bands into a low and a sub-low range. It enhances the lows and sub-lows by using an integration method, rather than the usual filter method. The end result of the integration means there is less phase change created by the device. The two low outputs are sent to 4 separate Bryston 7's running in mono mode. Each Bryston 7 powers a driver (only 2 of the 4 available ServoDrive speakers are currently used).

The final 2 outputs, for the sub-lows are sent to the 8 inputs of four Bryston 4 amplifiers running in stereo mode. This provides 8 outputs to drive the 8 sub-low frequency speakers.

3 Noise Chamber System Components

This section deals with elements the operator should understand before trying to use the system. It covers the control room console, system safeties, powering on and off the system, as well as the intercom system available in the noise chamber.

3.1 The Control Room Console

Before someone tries to run the system, it is important they understand how to use the operator console (located in the Control Room). Ultimately, this is where the noise room is controlled from and understanding how the console works goes a long way to ensure no hearing damage is done in the room.

3.1.1 Power Switch

The power switch has two settings, remote and local. It should normally be set to remote, meaning power is delivered from the rack unit after the safety key has been activated.

The local mode can be used, but is not recommended. It powers the console regardless of the state of the rack. It's not a problem to run locally, but it does remove one level of safety from the system.

3.1.2 Attenuator Enables

The level presented in the room is ultimately controlled by the 3 attenuators in the system. Each attenuator works in parallel with each other, meaning if any one of the attenuator is turned up, the volume is turned up. If any one is subsequently turned down, the level comes down.

There are 3 toggle switches used to enable or disable the three attenuators. Normally all attenuators are enabled, however it is sometimes useful to disable the attenuator in the noise room, or the rack unit, to ensure no one but the operator controls the level.

Attenuator A is the main control room unit, attenuator B is in the noise chamber and attenuator C is in the rack unit.

3.1.3 LEDs

The various LEDs on the top right of the operator console display which kill switch was activated. In practice it is never used (there is generally only one subject run at a time), but if you had multiple subjects, it could be used to determine who stopped the noise.

3.1.4 Kill Push Button

This is the main kill switch for the system. It works similar to a toggle switch. Hit it once and it stops the noise to the system. Hit it again and it re-activates the attenuators which control the system (note: the system does not automatically start making noise, the attenuators become active, but are in a state of fully attenuating). If the system is killed by a subject, the kill switch on the console first clears the subject's kill switch and then activates itself. The rule is that if the red LED on the kill button is on, no sound will come out. If the green LED is on, sound will come out.

3.1.5 Mic Level

This dial controls the level of the operator's microphone sent to the intercom system (if the subject can't hear the operator talking, try turning up this volume).

3.1.6 Comm Level

This dial controls the level of the intercom in the operator's ear (if the operator can't hear the subject speaking, try turning up this volume).

3.1.7 Push to Talk

If the subject is connected to the intercom system, the operator must push the 'push to talk' button while speaking. This will allow the subject to hear the operator.

3.1.8 Talkback Selector

The Talkback Selector gives the operator the opportunity to select which of the 4 subjects (or all) they wish to talk to. In practice, it is always left at 1. There has never (to my knowledge) been more than 1 subject in the room at a time.

3.1.9 Source Level

This control is never used.

3.1.10 Source/Monitor Switch

This switch is left in the Monitor position.

3.2 Built-in System Safeties

Before talking about the system and how to make it work, it is appropriate to outline some of the safety features built into the system. These safety features are important because the system has the potential to damage an individual's hearing, if used incorrectly.

3.2.1 Power up State

When the system is first powered up, by default, the volume controls (attenuators) will be in the fully attenuating mode and 4 mute switches on the frequency dividing system will be 'on'. That means there is no way for the system to power up with noise on.

3.2.2 Safety Key

On the master console (in the control room), a safety key exists. The key should be left in the 'on' condition (i.e. the safety is on) when the system is not in use. When the safety is on, the volume

control will be in the fully attenuating position. This key is under human control and should be used appropriately to ensure unauthorized use of the system does not occur. The operator should turn the safety on and remove the key if they are leaving the system unattended. This ensures no noise will be heard from the system.

3.2.3 Kill Switches

The kill switches (which immediately turn the noise off) will power up in the 'on' state (i.e. the system will be 'killed').

3.2.4 Video Link

As an added safety for subjects, a one-way video link can be established between the noise chamber and the control room. This allows the operator to watch the subject in the noise chamber. If the subject experiences a problem, they can signal via hand motions and (provided the operator is watching) the operator can react appropriately.

To establish the video link, the camera must be plugged in (it will automatically be powered on). The operator must turn on the monitor in the control room.

3.3 Powering On the System

It is important the system be powered up properly and this section describes how to do that. The reason for the given sequence is to 1) ensure the safety of anyone in the noise chamber and 2) ensure the equipment used in the system is treated properly.

3.3.1 Power Bar On

The first step is to power up the processing portion of the system. This part of the system handles generating and processing of the signal before it is delivered to the amplifiers.

The power bar is located roughly in the middle of the rack on the right side. It is a red rocker switch, located on a grey rack-mounted power bar and will light up when the system is on.

3.3.2 Configure the System

The second step in the power up procedure is to configure the system. This step is covered in sections 4.3 to 4.8 with more detail in the Appendices. Please refer to those areas for details on how to prepare the system to run the noise you desire.

3.3.3 Mute Switches

The frequency dividing unit has 4 mute switches (one for each of its frequency bands). These switches must be de-activated (un-muted) in order for them to pass any signal through. The LEDs above the switches will go out when the mute has been de-activated.

It has been noted recently that one of those switches seems to be a bit sticky and sometimes requires a couple of pushes before the LED goes out.

3.3.4 Release the Safety

At this point, the operator must go into the control console to continue. Please note that up until now, it is impossible to have any noises being generated in the room. Even with the amplifiers on and the noise source generating, the kill switches are still active so no sound will be presented. This ensures the operator must go into the control console to do the final activation of the system. The idea here is the operator will have to ensure anyone in the room has proper hearing protectors on before any noise is generated. It is a safety feature of the system.

To release the safety, turn the key from the 3 o'clock (on) position to the 12 o'clock (off) position. It has been noted recently that the safety key is somewhat contrary to turn. If the key is pushed down slightly while it is turned, it usually turns easily.

3.3.5 Releasing the Kill Switches

When the system is powered up, the kill switches are active (i.e. the system is killed). At this point in the power-up procedure, we are ready to activate the volume controls and the final step in doing that is to release the kill switches. On the operator's console, simply click the 'kill' button until the light goes green. A red LED means the system is stopped (killed). The green LED indicates the volume switches are active.

The system is setup such that when the kill switches are de-activated, the volume will be at its minimum (i.e. fully attenuated). There is no risk to clearing the kill switches—no noise will be heard upon doing this.

Once the green LED is on, the volume controls are active. It is up to the operator to ensure that all people in the room have adequate hearing protection for noise to be generated.

The noises in the system are generally setup such that the volume control is dialed up fully. It is possible to control the volume by not turning the dial all the way up. Each LED on the volume dial represents 3dB and each LED requires two indexed positions on the dial (the operator can feel the indexes when the dial is turned).

3.4 Intercom System

The noise chamber has a built-in communication system which can be used to communicate with subjects while the noise is running. It can also be used to inject test stimuli (like DRT/MRT words) to headsets under test.

The intercom system has an amplifier module in the chamber which can be used to mix two signals into one and deliver the mixed signal to the device under test (usually a headset worn by a subject or a manikin). The two signals being mixed are usually:

- the talkback microphone used by the operator to talk to the subject. This signal is only active when the microphone is keyed, using the 'push to talk' button on the console.
- a signal being presented to the subject. Typically, this would be a DRT or MRT, but could also come from a DAT player, or any other signal source. This signal must be connected to the '1' connector on the box in the noise chamber. The other end of the cable (in the control room) must be connected to the source of the signal.

In order for this system to function, the amplifier in the noise chamber must be powered using the power supply located on the floor of the chamber. A small toggle switch on the power supply will turn it on. This powers the amplifier delivering sound to the subject.

3.5 Powering Off the System

Equally important for the longevity of the system is the proper shutdown of all components of the system. This is especially important if the system has been running for some length of time and the amplifiers are producing heat.

3.5.1 Safety On in the Control Room

The first step is to place the system in the safe mode, by enabling the safety key in the control room. This ensures that no sound will be produced during the shut down procedure.

3.5.2 Turn off the Amplifiers

The amplifiers should be turned off and unlike when they are started, they can be turned off as quickly as you'd like. Watch for and ensure the LEDs on the front of each amplifier goes out. The Bryston 7's have LEDs which will glow for some time (15 to 20 seconds) after the power is switched off, but they will go off.

The buttons on the amplifiers have all recently been replaced, however, they occasionally still become sticky. Please ensure the LED goes off when the switch is hit—this guarantees the amplifier is off.

3.5.3 Stop and Rewind the Tape

If the DAT drive has been used as the source of the noise, now is a good time to stop the tape and rewind it to the beginning. This ensures the tape is ready to go the next time it is needed. If the noise generator has been used as the source, there is nothing to be done.

3.5.4 Power off the Power Bar

The final step in stopping the system is to turn the power bar off. See section 3.2.1 for the location of the main power bar. A note here before the power bar is turned off: if the system has been running for some time, it will have generated some heat and it will continue to be warm for a while after the amplifiers are turned off.

4 Configuring the System

For day to day operation of the noise chamber, it is a relatively simple matter of configuring the system to run a particular noise into the room. All attempts have been made to simplify the procedure, however there are six things which need to be checked in order to be sure the proper level and spectrum is presented. Those six items are presented here.

For a complete listing of what the settings should be, refer to the Appendices, where each noise source is listed. Along with each noise source, you will find; the programming values for the DEQ7, all of the settings for the six items presented in this chapter, and a graphical representation of the spectrum presented at the 4 prime locations in the room.

4.1 Input Source

The most important element of the system is the origin of the noise. There are currently 2 ways to feed the system a noise. The first method is from the analog output of the DAT drive. This uses the output from the DAT drive (the left channel is used). The cable is marked with a yellow band at each end. The output of the DAT drive has 2 settings, selectable by a switch on the back of the unit. The notes in Appendix A indicate which of the 2 settings should be used.

The second option for feeding the system is from the B&K Noise Generator. This generator can be configured to create many different continuous noises. The most common used at DRDC Toronto is Pink noise from 20Hz to 20kHz. In order to use this noise source, the cable to the Graphic EQ must be changed to the blue cable, using the TRS input.

It should be noted that the TRS (blue) cable will override the signal being sent on the balanced (yellow) cable, if they both happen to be plugged in. It is best to only plug in one of the two cables so there is no confusion.

4.2 Graphic EQ Input Sensitivity

On the back of the Graphic EQ, the input sensitivity can be set via a small slider switch. The two possible settings are +4dB or –20dB. This setting can be used to drop the level of the input to the Graphic EQ. If this level is too high, the Graphic EQ will distort the input (the LEDs on the front of the unit will indicate the level is high). It may or may not be possible to hear the distortion.

4.3 Digital EQ Program

The most important element in the chain used to modify the spectral content of the noise input is the DEQ7. By using the DEQ7's ability to save filter settings, the system can be configured quickly to handle different noise sources. It is essential the proper program be loaded when listening to a particular signal.

A very common mistake made by the operator is to 'overwrite' a program in the DEQ7. It is a simple matter to do and it has been done on numerous occasions by many operators of the system. The only solution is to re-program the overwritten program with the values from the Appendices. This takes some time so try to avoid making this mistake!

4.4 Digital EQ Input Level

The input level on the DEQ7 can be used to lower the overall level getting into the DEQ. This again can be handy if the level off the Graphic EQ is too high. For most situations, we try to use the DEQ7 at the 'Yellow' marker, but in cases where the level of the sound is not too high (like the Ops Room noise), we needed to lower the level at this stage (to the green marker).

4.5 ELF Cutoff Frequencies

The ELF cutoff frequency sometimes need to be modified to control the lower end of the spectrum. The amount of energy at the low frequencies is very difficult to control and the cut-off frequency used on the ELF is one of the methods used to try and control the low end. This is done with small slider switches on the front of the unit. What settings are used are determined when the room is tuned (Section 7).

4.6 Amplifier Selection

One easy way to control the level of the signal getting into the room is to reduce the number of amplifiers actively driving the room. There are situations where too much low frequency power is available and the solution is to turn off some of the sub-low drivers. The system currently uses 2 settings of amplifiers (either the yellow amps or the blue amps). The blue amplifiers only include one of the sub-low amplifiers. The yellow amps include all four.

When the amplifiers are being turned on, you should do it in a slow progression. It is known that if all amplifiers are turned on at the same time, a circuit breaker will blow, due to the large power-on current required by the amplifiers. By individually turning on the amplifiers at a rate of one per second (or slower), the system will power up properly without tripping breakers.

One other thing to watch for with the amplifiers: there are LEDs on the fronts which indicate they are turned on. Occasionally the switches are sticky (in particular the switches for the 4B's). Please ensure the LEDs are active after you've hit the on button.

5 Recording New Sources

In the past, two slightly different methods to record the noise sources for the noise chamber have been used. Fundamentally, they are in fact the same, however each uses different equipment and the methods are slightly different for that reason.

It should be noted at this point that one of the fundamental requirements for recording a noise source is that all equipment must be portable (and battery powered). Extension cords and 120V power are rarely available for these recordings since many of the noise environments involve moving vehicles (airplanes, helicopters, tanks, etc...). This puts limits on what equipment is available for use.

Common to both methods below are the following general guidelines:

- Try to get 5 minutes of steady state noise for each noise condition. This gives a good length of time to measurement the spectrum and also gives an indication of how consistent the noise field is.
- Due to the amount of setup involved with measuring a noise source, try to be as complete as possible with the noise field of interest. By this, it is implied that all possible conditions of noise should be measured. As an example, if a helicopter is being measured, all flight conditions should be obtained. This includes (if possible):
    - Engine startup
    - Engine shutdown
    - Take off
    - Landing
    - Hovering in ground effect
    - Hovering outside ground effect
    - Transition to forward flight
    - steady forward flight (at different speeds)
    - doors open and/or closed (in the various flight conditions above)
- It is also wise to record at least two different locations in the aircraft/vehicle. The spectrum changes in the various areas and having two recording gives an option to choose which area is of more interest.
- It is best to talk to the crew involved to find out what noises are of interest and how each should be obtained.

5.1 B&K Microphone

The first method uses B&K microphones, with a portable power supply. These microphones can be directly calibrated with a pistonphone or other calibrator. The output of the microphone is directly coupled to a portable DAT drive. The record level on the DAT drive can be used to adjust the level going to tape. Once set, the record level is not adjusted (we usually tape the control in place). The calibration tone is printed to the tape both before and after recording the noise of interest. This is for later use to get the complete spectrum of the noise (see the section on tuning the noise chamber).

This method has the advantage of using slightly less equipment and being able to directly calibrate the entire system with the calibration tone on the tape. The drawback is that the cable lengths involved are fairly small. We have long B&K cables to use in the system, but they are quite large and bulky and difficult to place in unobtrusive locations. For large aircraft (like the Hercules) this can be a problem.

5.2 AKG Microphone

The second method involves using AKG C414 microphones. These are standard, studio grade microphones that use balanced XLR cables. Their output is flat for the audio frequency range. They are condenser microphones and as such, require phantom power to operate. Our Tascam portable DAT drives have the option of providing phantom power so this is not a problem. The output of the microphone is directly coupled to the DAT drive and as in the previous method, the record level is adjusted using the DAT drive's record level.

Unfortunately with this method, it is not possible to print a calibration tone on the tape since there are no calibrators to couple to the microphone. When this method is used, it is necessary to also use a portable sound level meter and take measurements of the spectrum of the noise, while doing the measurements. The noise spectrum can be calibrated using the sound level meter measurements at a later point.

This method has the advantage of being able to use relatively unobtrusive cabling for long lengths. This works well in large aircraft where you want to monitor distant locations, but have control of the tape decks at a central spot.

6 Creating Noise Loops

Creating a noise tape for the system is as much an art as it is a science. The basic idea behind this task is to take the recordings made in the previous section and loop a 'typical' segment. This looped noise can be recorded to a DAT for use as input to the system. When the noise loop is being made, no thought of the spectrum of the noise is taken into account. That is done in the next section, when the noise chamber is tuned.

The first step in creating the noise loop is to get the noise onto a computer. The computer is used as a tool for creating the noise loop. This transfer to computer is done directly from the DAT into the computer via a SPDIF input. This is a digital transfer of information so no information is lost (i.e. we don't use analog since it relies on both a digital to analog and another analog to digital conversion). The computer called 'pcaudiolab' has a SPDIF input that can be used to get the digital data from the DAT to the computer.

Once a recording has been done on the computer, it is possible to use any of a number of software packages to create the noise loop. CoolEdit is one such program which works well for this application (GoldWave is another). They are simple waveform editors which will let the user trim and loop the noise file. As mentioned earlier, at this point in time, no concern is given to the spectrum of the noise. At this point we are simply looking for a good loop (no audible clicks or thumps at the loop point).

Here is where the art comes into play. There is no tried and true method for getting the perfect loop. It simply takes time and some educated guesswork. In order to create a loop that doesn't cause any irregularities when the loop goes from the end to the beginning again, it is easiest to start and end the loop at a zero crossing (where the waveform crosses the axis). This can be done visually.

The best results are obtained if a stretch of continuous noise of 10 to 20 seconds long can be found. Once looped properly, it will be virtually impossible to hear the loop point and it also gives a truer representation of the actual noise (and its subtle variations).

Most of the tweaking of the loop point can be done with careful listening on any set of speakers (or headphones). Once a good loop is obtained, it should be auditioned on the noise system. The noise system has a much better low frequency response than most speaker systems so an apparently good loop can suddenly appear to have a 'thump' at the loop point. If this turns out to be the case, try again.

Once a good loop has been found, record a full tape of the sound. To do this, simply play the signal from the computer to a DAT drive (use the digital output from the computer to the digital input of the tape deck) and record. Use the full tape to give the maximum length of noise.

7 Tuning the Noise Chamber to New Noises

Once a full DAT has been recorded, the room needs to be fine tuned to achieve the proper spectrum. The DAT will give a good approximation however, due to the acoustics of the noise chamber, the actual spectrum and level at the listening positions will not be what was measured at the source. The spectrum can be modified using the DEQ7 to achieve a spectrum more closely representing the measurement.

The first thing to obtain is the spectrum of the original noise. Depending on how the source was recorded, there are different techniques to obtain the spectrum. If the B&K method was used (from section 5), then the original tape needs to be played into the B&K Spectrum Analyzer (B&K 2133). The analyzer should be calibrated to the calibration tone recorded on the tape. The portion of the tape used to create the loop would be the best segment to analyze. Use at least a 32 second sample time to get the spectrum from the tape. The spectrum can be saved to disk and transferred to Excel using the program (nextract) written for the Mac by Steve King.

If the AKG microphones were used, then the spectrum must be taken from the sound level meters. It is easiest to write the numbers down from that meter.

What has been found to be best is to use the dual channel display on the analyzer. One channel is the 'live' microphone, positioned in the noise chamber at one of the four positions (the red dots). The other channel is the desired spectrum. This tracing must be manually entered into the analyzer. See the manual for the analyzer on how to do this.

Now you can use the analyzer in its 'oscilliscope' mode, where it uses a rolling window to display the spectrum in real time. You now simply use the DEQ7 to adjust the parameters on the various filters to achieve levels for each spectral band. Here again, it is more art than science. What has worked in the past is to start with a basic (pre-programmed) filter from the DEQ7's arsenal. Then adjust from that point. It is difficult to know which of the pre-programmed settings is best to use. In some instances a Graphic EQ is best, while in others a parametric is. It is simply trial and error.

8 Constants

This portion of the document contains information about the setup of various pieces of equipment which are involved in the noise chamber, but whose settings never change. If the values ever change, this section could be used to reset the settings back to their proper value.

8.1 Graphic EQ

The graphic EQ has sliders for each of the frequency bands on the front panel of the device. During normal use, these sliders never change, however the setting of those sliders could be inadvertently changed. It is impossible to get the sliders at exactly the same position, but it is possible to get very close.

Frequency Band	Value
20Hz	+9
25Hz	-7.5
31.5Hz	+1
40Hz	-10
50Hz	+9
63Hz	+9
80Hz	+1
100Hz	+8
125Hz	-5
160Hz	-6
200Hz	+2
250Hz	-3
315Hz	-4.5
400Hz	+4
500Hz	+2.5
630Hz	0
800Hz	+6
1kHz	+9
1.25kHz	-2.5
1.6kHz	+4.5
2kHz	-2.5
2.5kHz	+6
3.15kHz	+4
4kHz	+2
5kHz	+7
6.3kHz	+7.5

8kHz	+12
10kHz	+12
12.5kHz	+7
16kHz	+6
20kHz	0

The output level of the Graphic EQ (set on the back of the unit) is always +4dB.

8.2 DEQ7

The input level of the DEQ7 (set on the back of the unit) is always +4dB for both inputs.

8.3 Frequency Dividing System

These settings are for the BSS Frequency Dividing system used to create the 4 separate audio bands. The first four sections are found on the front display panel.

8.3.1 Band 1

Level: 2.0 dB
Normal Polarity
Not mono low
Phase = 0

8.3.2 Band 2

Level: 0 dB
Normal Polarity
Phase = 0

8.3.3 Band 3

Level: 2.2 dB
Reverse Polarity
Phase = 0

8.3.4 Band 4

Level: 0.0 dB
Normal Polarity

8.3.5 Rear Panel Settings

On the back of the unit, there are small slider switches to control the 'limiter threshold levels'. All those switches are off (down).

The mode switch is set to 3-way/4-way.

8.4 Digital Attenuators

These settings are found on the back of the unit.

Preset: 0
Priority: 0

The attenuators themselves are daisy chained together. The output of one is connected to the input of the next. That continues until all four channels are connected. This ensures all signals are attenuated the same amount by the same control.

8.5 Delta Max

These settings are for the Electro Voice DeltaMax units which divide the frequencies for the high and low drivers of those speakers. There are 2 units and each have the same settings.

Output Levels:
LF = 0
HF = +5

Amplifier Calibration:
LF = 800
HF = 400

8.6 ELF

The ELF device has many small configuration switches on the front panel. Up to five of them may be used to configure the system (typically only three are used). The other switches are rarely used. In particular, the only switches in the 'up' position are the '2' switch on the left and the polarity switch towards the right. All other switches are in the down position (with the possible exception of the switches used to control the low frequency cut off).

8.7 Bryston Amplifiers

The Bryston amplifiers have volume controls recessed on the front panel. They can be used to set the volume for the channels of the amplifier.

The volume settings are not all identical and their approximate locations are described here.

On the back of each Bryston, there are a few switches used to configure the amplifier. All of the grounds are lifted and the polarity is always normal. Some of the amplifiers run in stereo mode (the highs and the sub lows), while some run in mono mode (the mids and the lows).

9 Noise A: Mauve Noise

This noise is flat at 85dB up to 1kHz, then it increases at 3dB per octave (like white noise). (i.e. pink noise up to 1kHz, white noise from 1kHz to 20kHz).

9.1 DEQ Program #40

This noise source was created using the Parametric EQ (4+4) L = R program on the DEQ 7 (original program #13)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	50
R Out Level	70
L Band 1 (F)	63 Hz
L Band 1 (G)	+3.0
L Band 1 (Q)	10
L Band 2 (F)	112 Hz
L Band 2 (G)	+1.0
L Band 2 (Q)	2.2
L Band 3 (F)	3.4 kHz
L Band 3 (G)	+3.4
L Band 3 (Q)	1.1
L Band 4 (F)	12 kHz
L Band 4 (G)	+13.0

L Band 4 (Q)	1

9.2 Configuration

1) Input from B&K Noise Generator, unbalanced cable, 20-20KHz Pink Noise, 1.25V.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 40, "Patti's Noise".
4) Digital EQ input level at the GREEN arrow.
5) ELF cutoff –8-4-2- (none, 22Hz), both channels.
6) Power on all amps with BLUE dots.

9.3 Spectrum

10 Noise B: Sea King

This noise is from a recording of the Sea King. The recording was looped on a computer and a 2 hour tape was created. It was recorded at the Senso position.

10.1 DEQ Program #41

This noise source was created using the Parametric EQ (4+4) L/R program on the DEQ 7 (original program #14)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	100
R Out Level	70
L Band 1 (F)	20 Hz
L Band 1 (G)	-5.4
L Band 1 (Q)	1
L Band 2 (F)	32 Hz
L Band 2 (G)	-3.0
L Band 2 (Q)	1.0
L Band 3 (F)	63 Hz
L Band 3 (G)	-9
L Band 3 (Q)	1.6
L Band 4 (F)	125 Hz
L Band 4 (G)	+8
L Band 4 (Q)	1
R Band 1 (F)	250 Hz
R Band 1 (G)	+1
R Band 1 (Q)	1
R Band 2 (F)	500 Hz
R Band 2 (G)	+3
R Band 2 (Q)	1.0
R Band 3 (F)	2 kHz
R Band 3 (G)	+4
R Band 3 (Q)	0.5
R Band 4 (F)	
R Band 4 (G)	0
R Band 4 (Q)	

10.2 Configuration

1) Input from DAT, balanced cable, -10dB output.
2) Graphic EQ input -20dB sensitivity.
3) Digital EQ Program 41, "Sea King Hover".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff -2- (none, 10Hz), both channels.
6) Put "Sea King" tape into DAT machine.
7) Power on all amps with Yellow dots.

10.3 Spectrum

11 Noise C: Leopard Tank

This loop is approximately 4 seconds in length.

11.1 DEQ Program #43

This noise source was created using the Parametric EQ (4+4) L/R program on the DEQ 7 (original program #14)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	80
R Out Level	100
L Band 1 (F)	20 Hz
L Band 1 (G)	-14.0
L Band 1 (Q)	2.5
L Band 2 (F)	32 Hz
L Band 2 (G)	+18
L Band 2 (Q)	10
L Band 3 (F)	63 Hz
L Band 3 (G)	+4.6
L Band 3 (Q)	0.7
L Band 4 (F)	125 Hz
L Band 4 (G)	+5.6
L Band 4 (Q)	8.0
R Band 1 (F)	500 Hz
R Band 1 (G)	+8.0
R Band 1 (Q)	0.32
R Band 2 (F)	250 Hz
R Band 2 (G)	+3.8
R Band 2 (Q)	3.2
R Band 3 (F)	2 kHz
R Band 3 (G)	+2
R Band 3 (Q)	1
R Band 4 (F)	4 kHz
R Band 4 (G)	+4.4
R Band 4 (Q)	1

11.2 Configuration

1) Input from DAT, balanced cable, +4dB output.
2) Graphic EQ input -20dB sensitivity.
3) Digital EQ Program 41, "Leo Tape Loop".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff –0- (none 8Hz), both channels.
6) Put "Leo Cruise Loop" tape into DAT machine.
7) Power on all amps with Yellow dots.

11.3 Spectrum

12 Noise D: Ops Room

This tape was made during simulated war game on board the Halifax class frigate, the HMCS Ville de Quebec CPF, from March 28 to 31, 2000. The frigate was off the coast of St. John's, Newfoundland in the North Atlantic.

12.1 DEQ Program #44

This noise source was created using the Parametric EQ (4+4) L = R program on the DEQ 7 (original program #13)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	100
R Out Level	100
L Band 1 (F)	20 Hz
L Band 1 (G)	-5
L Band 1 (Q)	.56
L Band 2 (F)	320 Hz
L Band 2 (G)	-8.5
L Band 2 (Q)	7
L Band 3 (F)	950 Hz
L Band 3 (G)	-4.4
L Band 3 (Q)	2.2
L Band 4 (F)	10 kHz
L Band 4 (G)	-4.4
L Band 4 (Q)	0.1

12.2 Configuration

1) Input from DAT, balanced cable, +4dB output.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 44, "Ops Room".
4) Digital EQ input level at the GREEN arrow.
5) ELF cutoff -0- (none, 8Hz), both channels.
6) Put "Ops Room Noise Loop" tape into DAT machine.
7) Power on all amps with BLUE dots.

12.3 Spectrum

13 Noise E: Griffon Noise

The Griffon loop was taken from a tape recorded on June 5, 1998. The flight was out of CFB Borden. Altitude was 1500ft and the air speed was 120kts. All doors were closed. The recording was made with the B&K microphone setup, with the ambient microphone hanging from the ceiling, 1 foot behind the copilot's head.

13.1 DEQ Program #40

This noise source was created using the Parametric EQ (4+4) L = R program on the DEQ 7 (original program #13)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	100
R Out Level	100
L Band 1 (F)	20 Hz
L Band 1 (G)	-6.0
L Band 1 (Q)	6.3
L Band 2 (F)	50 Hz
L Band 2 (G)	-10
L Band 2 (Q)	2.8
L Band 3 (F)	2 kHz
L Band 3 (G)	-4.2
L Band 3 (Q)	0.45
L Band 4 (F)	8 kHz
L Band 4 (G)	-12
L Band 4 (Q)	2.8

13.2 Configuration

1) Input from DAT, balanced cable, +4dB output.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 45, "Griffon Loop".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff – 8 - 4 – (20 Hz), both channels.
6) Put "Griffon Noise Loop" tape into DAT machine.
7) Power on all amps with YELLOW dots.

13.3 Spectrum

14 Noise F: Speech Noise

This noise was created for the round-robin testing performed at DRDC Toronto. A set of ANR headsets were tested in this noise.

14.1 DEQ Program #46

This noise source was created using the Graphic EQ program on the DEQ 7 (original program #??)

Parameter	Value
L Delay	0

R Delay	0
L Out Level	54
R Out Level	100
L 40Hz	+4
L 50Hz	0
L 63Hz	+9.0
L 80Hz	+2.8
L 100Hz	+5.0
L 125Hz	+9.5
L 160Hz	+3.8
L 200Hz	+5.0
L 250Hz	+12.0
L 320Hz	+3.8
L 400Hz	+5.4
L 500Hz	+8.5
L 630Hz	+4.6
L 800Hz	+7.0
L 1.0kHz	+5.0
L 1.25kHz	+1.8
L 1.6kHz	+2.6
L 2.0kHz	+2.0
L 2.5kHz	+1.4
L 3.2kHz	0
L 4.0kHz	-0.6
L 5.0kHz	-4.4
L 6.3kHz	-3.0
L 8.0kHz	-6.5
L 10kHz	-4.0
L 13kHz	0
L 16kHz	0

14.2 Configuration

1) Input from B & K Noise Generator, unbalanced cable, 20-20KHz Pink Noise.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 46, "Speech Noise".
4) Digital EQ input level at the RED arrow.
5) ELF cutoff – 8 - 4 –2 – (22 Hz), both channels.
6) Power on all amps with BLUE dots.

14.3 Spectrum

15 Noise G: Aurora Noise

This recording was made on a Nagra tape recorder in 1981. It was made at the NASO position of the aircraft.

15.1 DEQ Program #47

This noise source was created using the Graphic EQ L = R program on the DEQ 7 (original program #13)

Parameter	Value
L Delay	0

R Delay	0
L Out Level	100
R Out Level	100
32 Hz	-4.6
63 Hz	-0.6
125 Hz	1.4
250 Hz	-5.4
500 Hz	-3.2
1k Hz	-1.4
2k Hz	-1.6
4k Hz	-2.4
8k Hz	0.8
16k Hz	0

15.2 Configuration

1) Input from DAT, balanced cable, -10dB output.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 47, "Aurora Navcom".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff – 2 – (10 Hz), both channels.
6) Power on all amps with YELLOW dots.

15.3 Spectrum

16 Noise H : Pink Noise (80dB)

This noise is pink noise (equal energy per octave) at a level of 80dB in each 1/3 octave frequency band. It has been used for various studies.

16.1 DEQ Program #48

This noise source was created using the Parametric EQ (4+4) L = R program on the DEQ 7 (original program #13)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	100
R Out Level	100
L Band 1 (F)	40 Hz
L Band 1 (G)	-18
L Band 1 (Q)	4.5
L Band 2 (F)	21 Hz
L Band 2 (G)	+18
L Band 2 (Q)	9
L Band 3 (F)	40 Hz
L Band 3 (G)	+10.5
L Band 3 (Q)	0.9
L Band 4 (F)	45 Hz
L Band 4 (G)	-18
L Band 4 (Q)	3.2

16.2 Configuration

1) Input from B & K Noise Generator, unbalanced cable, 20-20KHz Pink Noise.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 48, "Pink Noise".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff – 8 - 4 - 2 – (22 Hz), both channels.
6) Power on all amps with YELLOW dots.

16.3 Spectrum

17 Noise I: Hercules Noise (Flight Engineer)

This recording was made on January 16, 2002, somewhere over Southern Ontario. The Hercules was flown out of CFB Trenton with the standby Search and Rescue crew. The aircraft was at 8000ft, travelling at 240kts. The AKG C414 microphones were used, hanging from the ceiling slightly behind the Flight Engineer's seat (with the seat all the way forward).

17.1 DEQ Program #49

This noise source was created using the Parametric EQ (4+4) L = R program on the DEQ 7 (original program #13)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	100
R Out Level	100
L Band 1 (F)	22 Hz
L Band 1 (G)	+8
L Band 1 (Q)	1
L Band 2 (F)	63 Hz
L Band 2 (G)	-7
L Band 2 (Q)	7
L Band 3 (F)	40 Hz
L Band 3 (G)	-3.4
L Band 3 (Q)	10
L Band 4 (F)	5.6 kHz
L Band 4 (G)	+6.5
L Band 4 (Q)	0.1

17.2 Configuration

1) Input from DAT, balanced cable, +4dB output.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 49, "Hercules".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff – 8 - 4 – (20 Hz), both channels.
6) Put "Hercules Noise Loop" tape into DAT machine.
7) Power on all amps with Blue dots.

17.3 Spectrum

18 Noise J: Hercules Noise (Load Master)

This recording was made on January 16, 2002, somewhere over Southern Ontario. The Hercules was flown out of CFB Trenton with the standby Search and Rescue crew. The aircraft was at 8000ft, travelling at 240kts. The AKG C414 microphones were used, horizontally centered in the aircraft, 6 feet forward of the ramp hinge, 6 feet off the floor of the aircraft. The microphone was held in place by large rubber bands.

18.1 DEQ Program #50

This noise source was created using the Parametric EQ (4+4) L = R program on the DEQ 7 (original program #13)

Parameter	Value
L Delay	0
R Delay	0
L Out Level	100
R Out Level	100
L Band 1 (F)	63 Hz
L Band 1 (G)	1
L Band 1 (Q)	2.2
L Band 2 (F)	250 Hz
L Band 2 (G)	-5.2
L Band 2 (Q)	0.56
L Band 3 (F)	8 kHz
L Band 3 (G)	-3.4
L Band 3 (Q)	0.56
L Band 4 (F)	6.3 kHz
L Band 4 (G)	0
L Band 4 (Q)	1

18.2 Configuration

1) Input from DAT, balanced cable, +4dB output.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 49, "Hercules".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff – 8 - 4 – (20 Hz), both channels.
6) Put "Hercules Noise Loop" tape into DAT machine.
7) Power on all amps with Blue dots.

18.3 Spectrum

19 Noise K: Pink Noise 86dB

This noise is pink noise (equal energy per octave) at a level of 86dB in each 1/3 octave frequency band. It was created to compare against C. Giguere's results on his human head test fixture.

19.1 DEQ Program #51

This noise source was created using the Parametric EQ (4+4) L = R program on the DEQ 7 (original program #13)

Parameter	Value

L Delay	0
R Delay	0
L Out Level	50
R Out Level	70
L Band 1 (F)	63 Hz
L Band 1 (G)	+3.0
L Band 1 (Q)	10
L Band 2 (F)	112 Hz
L Band 2 (G)	+1.0
L Band 2 (Q)	2.2
L Band 3 (F)	3.4 kHz
L Band 3 (G)	+3.4
L Band 3 (Q)	1.1
L Band 4 (F)	12 kHz
L Band 4 (G)	+13.0
L Band 4 (Q)	1

19.2 Configuration

1) Input from B & K Noise Generator, unbalanced cable, 20-20KHz Pink Noise.
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Program 51, "Pink Noise 86dB".
4) Digital EQ input level at the YELLOW arrow.
5) ELF cutoff – 8 - 4 - 2 – (22 Hz), both channels.
6) Power on all amps with YELLOW dots.

19.3 Spectrum

20 Noise L: Standing Wave Demonstration

This 'noise' source is not noise at all, it is a pure 17Hz tone. The purpose of this source is purely demonstrative. Using a 17Hz tone, the room's primary mode is excited, even though it is out of our audible hearing range. The effect in the room is to experience almost nothing along the center line of the room, but maximum pressure at either end of the room. It is a great way to have people 'experience' a standing wave.

Please note: do NOT play this signal at full volume. Damage will result in the sub-low speakers, due to the amount of energy they are attempting to generate. Since humans can't hear frequencies this low, it is easy for the operator to turn the volume too high without noticing anything. By the time the speaker drivers start to make noises (from excess cone excursion), it is usually too late. One third to one half volume is plenty to get the idea of a standing wave.

20.1 DEQ Program

The program number is irrelevant for this noise source. The DEQ7 must be set to the 'bypass' position (i.e. no equalization done).

20.2 Configuration

1) Input from B & K Noise Generator, unbalanced cable, 17Hz Pure tone, 125mV
2) Graphic EQ input +4dB sensitivity.
3) Digital EQ Bypassed

4) Digital EQ input level at the YELLOW arrow.

5) ELF cutoff – 0 – (8 Hz), both channels.

6) Power on only the sub-low amplifiers (the bottom four amplifiers).

20.3 Spectrum

21 Reverb Time Measurements

Location: B104
Date: March 31, 2004
Performed by: Garry Dunn
Notes:
- all amps on
- DEQ7 bypassed
- ambient level in the room: 54 dBL (no air conditioning)
- ambient level in the room with noise on (linear): 94 dBL
- 5 repeats for each octave frequency

Category	31.5Hz	63Hz	125Hz	250Hz	500Hz	1KHz	2KHz	4KHz	8KHz	16KHz
1-5	2.40	2.03	1.09	1.45	1.43	1.40	1.03	0.89	0.77	0.52
2-5	1.25	0.62	0.71	0.80	1.44	1.41	1.11	0.86	0.64	0.46
3-5	2.35	1.04	0.69	.94	1.00	1.19	1.22	1.07	0.69	0.46
4-5	2.16	1.02	1.12	1.39	1.44	1.50	1.32	0.93	0.67	0.44
5-5	1.83	0.63	1.02	0.77	1.14	1.06	1.30	1.09	0.69	0.39
6-5	1.75	1.03	1.04	1.06	1.23	1.27	1.11	0.84	0.69	-
1-10	1.82	1.32	0.90	1.12	1.44	1.40	1.07	0.87	0.71	0.49
2-10	2.25	1.05	0.91	1.16	1.22	1.34	1.27	1.00	0.68	0.45
3-10	1.79	0.83	1.03	0.92	1.18	1.16	1.20	0.47	0.69	-
1-15	2.00	1.24	0.83	1.06	1.29	1.33	1.12	0.44	0.70	0.48
2-15	1.91	0.90	1.06	1.07	1.27	1.28	1.24	0.95	0.68	-
1-20	2.04	1.19	0.90	1.14	1.33	1.37	1.17	0.94	0.69	0.47
1-30	1.96	1.07	0.45	1.07	1.28	1.30	1.18	0.95	0.69	-

Appendix B

The DRDC Toronto Noise Simulation Facility Sound Library

Noise Simulation Facility Sound Library

This document describes the various sounds available for demonstration in the Noise Simulation Facility at DRDC Toronto.

22 Leopard Tank Cruise

Date of Recording	November 1993
Speed	
Terrain	
Comments	Tape label says: Passenger/Turret

23 Hercules Flight Deck

Date of Recording	Jan 16, 2002
Air speed	240kts
Altitude	8000ft
Location within Aircraft	Hanging from the ceiling slightly behind the Flight Engineer's seat (seat all the way forward). AKGC414 Mic's.

24 Hercules Load Master

Date of Recording:	Jan 16, 2002
Air speed:	240kts
Altitude:	8000ft
Location within Aircraft	Horizontally centered, 6 feet forward of the ramp hinge, 6 feet off the floor of the aircraft. Held in place by large rubber bands. AKGC414 Mic's.

25 Griffon

Date of Recording	June 5, 1998
Air speed	120kts
Altitude	1500ft
Location within Aircraft	1 foot behind the copilot's head. Hanging from the ceiling. B&K Microphone.

26 Sea King

Date of Recording	
Air speed	
Altitude	
Location within Aircraft	Senso Position

27 Ops Room

Date of Recording	May 26, 1999
Speed	
Comments	This recording was made by Lisa Massel while onboard the ship. The tape is labeled 'Weds War'.

28 Aurora

Date of Recording	1981
Air speed	
Altitude	
Comment	Taken from Nagra tape
Location within Aircraft	NASO Position

Other tapes we have, but have not been looped and calibrated the room for:

1. Bell Jet Ranger

2. Twin Huey

Appendix C

The MRT/MLP Software System

MRT/MLP Documentation

This document provides the reader with a description of how the MRT/MLP software works in conjunction with the TDT system. It should be considered required reading before changes to the software are attempted.

In order to understand this document, there are three fundamental acronyms that must be made clear to the reader:

- MRT Modified Rhyme Test. This is a procedure during which the listener is presented (visually) with 6 words. Audibly, the listener will hear one of them.
- MLP Maximum Likelihood Procedure. This is a mathematical model used to choose the presentation level of each word.
- TDT Tucker-Davis Technologies. This is the hardware/software system used to present the words to the listener.

Each of these will be explained in further detail later in this document.

29 Overview

As background, you must first understand what is involved with the MRT study. First and foremost, the listener is immersed in a noise field of interest. They are wearing a communication headset of interest. Into that headset, the program will inject an audio signal.

The basic concept is that the listener is presented with a carrier phrase, followed by a word. The carrier phrase is clearly audible, but the word may or may not be. Visually, the listener is given a choice of six words they may have heard. They are to choose (using a button box), which word they think they heard. If they heard nothing, they must still choose a word, before the next presentation is given to them. The implementation used at DRDC Toronto employs 50 repeats of the carrier phrase/word. After much experimenting, it was found that after 50 repeats, the system had reached its final level and further repetitions did not give any major level changes.

The level at which the word is presented is determined by the MLP procedure. The MLP procedure is designed to adjust the level of the word to the point where the listener is getting 60% correct. How it does this will be explained later.

The TDT system is used as the tool to present the words and the carrier phrase for the study to the listener. The TDT system is fully configurable in software to allow the exact presentation of audio required. In particular, the system has the ability to perform real-time attenuation of an audio signal. That is the fundamental requirement for a system used to present an MRT study.

The MLP portion of the system is a mathematically intense procedure and requires some computing horsepower to perform the calculations at a reasonable speed. This can be achieved with the computer used to run the TDT equipment.

30 MRT

MRT is the procedure of presenting an audible carrier phrase, followed by a word. Each word is similar to each other, but differs in either the initial consonant (raw, paw, law, jaw, thaw, saw) or the final consonant (teach, tear, tease, teal, team, teak). There are 50 sets of 6 words in total, 25 of which change the initial consonant, while the other 25 change the final consonant. The words are always preceded by a 'carrier' phrase, currently 'Next Word'. The carrier phrase is clearly audible and provides the listener with a warning that one of the 6 words is about to be presented. The group of 6 words is picked at random, as is the word within the group. No attention is currently paid to balance the number of words whose initial consonant changes with those whose final consonant changes.

31 MLP

The MLP is a mathematically intense procedure based on the assumption that the responses will fit a known psychometric curve. For the MRT procedure, it has been shown empirically that the responses will fall on the cumulative normal distribution function, with a standard deviation of 6dB. This is the psychometric function we use.

To understand what is involved with the math, it is necessary to realize the MLP procedure actually involves solving a family of curves. Each equation in the family is solved and the maximum of all the curves is used as the best guess for the next presentation. Based on the results of the listener's next answer, each curve is solved again and the new maximum is found. The curves being solved can be thought of as probabilities. After each trial, the software solves the basic question of:

> What is the probability of the subject getting the sequence of rights and wrongs so far, if the psychometric curve has a median of 'x'. ('x' varies from 0 to –30dB in 0.1dB increments.)

The answer can be determined by using the previous solution to the curve and multiplying by the probability of getting the answer that was received, based on each of the values at each point. In the current software, the family of curves covers a range of 30dB, in 0.1dB increments, thus giving 300 curves to be solved after each word is chosen.

32 Software

Each experiment we have performed with the MRT/MLP procedure has been different. This means the software is always changing to accommodate the different protocols. For this reason, it is necessary to examine the software to get a full understanding of the procedure/experiment. Unfortunately, this makes it more difficult to understand.

32.1 Directory Structure

All of the software is contained on the computer in the noise control room. It can be found in the C:\dciem\mrt directory. The experiments use the TDT equipment suite to present the signals to the listener. You should become familiar with that equipment first (the documentation is beside the system).

All programming is done in DOS. The TDT drivers do not work properly under the Windows environment (they look like they do, but they will fail). They do not work in a DOS window, within Windows either. Boot directly to DOS (hit F8 while the system is booting and select 'Command Prompt Only') in order for the system to work 100%.

The following directories are relative to the main directory for the MRT program.

32.1.1 M1

This directory contains the words for the male talker.

32.1.2 F1

This directory contains the words for the female talker.

32.1.3 HRTFS

This directory contains the HRTF data for the spatialized version of the MRT program.

32.1.4 NATO.GOD

This directory contains the session files and log files from the NATO round robin trials.

32.1.5 CEP

This directory contains the session files and log files from the CEP/SPH5 protocol.

32.1.6 EAGLE.CHR

This directory contains the session files and log files from the Christy Bell's Eagle headset testing.

32.1.7 EAGLE.ERI

This directory contains the session files and log files from the Erika Hasler's Eagle headset testing.

32.1.8 SAR

This directory contains the session files and log files from a SAR protocol.

32.1.9 NRC

This directory contains the software from NRC. They implemented a version of MLP for one of their studies. We didn't use any of this software—it is written in Pascal and is not documented well. It is very difficult to understand.

32.1.10 SIGNALS

This directory contains the other WAV files used during the experiment.

32.2 Development

No integrated development environment was used to create the software. It was done in standard text editors with manually created make files.

32.3 How the Protocols Run

All experiments run on the same basic premise. Each subject has an ID # associated with them. A session file is created (usually by hand) for each subject. That file defines what order of presentations the subject gets. The format of that session file is up to the programmer to determine, but would typically look like the following:

Quiet	CEP	On	m1	1	60%	Off
SeaKing	CEP	On	f1	1	60%	Off
Hercules	CEP	On	m1	1	60%	Off
Quiet	SPH5andCEP	On	f1	1	60%	Off
SeaKing	SPH5andCEP	On	m1	1	60%	Off
SeaKing	SPH5andCEP	On	f1	1	60%	Off
Hercules	SPH5	On	f1	1	60%	Off
Quiet	SPH5	On	m1	1	60%	Off

The file is tab delimited. Each column specifies a parameter of the MRT/MLP procedure. The columns above denote:

Column #	What it does
1	Background noise to be played
2	Headset to be worn
3	ANR on or off (always on in this example because that study did not involved ANR)
4	Talker (f1 is female, m1 is male)
5	# of repetitions (the system will run multiple

	versions of the MLP procedure if desired—we've never done this)
6	Target percentage (we've always used 60%--that is the point where the standard deviation amongst listeners is the least)
7	Feedback (determines whether or not the listener is given feedback as to the correct word spoken—never used)

As stated above, the session files are generally created by hand, but there is a 'genses' program which can be used to create session files, if so desired. This is an independent program which must be modified to generate the session (text) file. As a general rule, this file must ensure random-without-replacement for some of the variables, as well as (potentially) latin squares for others. Genses is a great idea if a large number of subjects are to be run, however, for small numbers, it's likely easier and quicker to manually generate the files.

Please note: each experiment requires its own setup. The above columns may or may not be sufficient to define the next experiment. It is up to the programmer to adjust the session files (and the MRT software) to handle what is required of the experiment. A note about timing as well: each line above results in approximately 4 minutes worth of time (during the experiment). It takes the average subject about 4 minutes to perform the entire MRT protocol. New subjects take a little longer. Each line represents one repetition of the entire MRT protocol. That implies 50 different repetitions of 6 words, each played at a different volume, depending on what the MLP math determines to be the volume. The final result of the 50 repetitions is the number we're looking for.

As the program is presenting each group of 6 words, it is also creating a file (called a .log file). The log file contains all the information needed to determine what the listener did (what words were presented, what the listener picked for the word, etc...). This log file is the file that gets analyzed after the study is performed.

Appendix D

Hearing Research Facility Backups

Hearing Research Facility Backups

This document outlines the backups that are in place for the Hearing Research Facility's system. After much experimenting, there was no complete backup made to the entire system, however, many of the components of the system have backups. You should be very comfortable with computers, if you want to attempt changing any of the components.

Let me first give you some background on the system. The entire system is run on an AST 286 computer. It uses a monochrome EGA monitor. The hard drive is an MFM type 31 drive. The system has a 1.2M floppy and boots to DOS 3.30.

The software has some dependencies on this hardware/software combination. As an example, there is a graphing part of the program which only seems to work with the current EGA video card in the system. Attempts were made to switch it to another EGA card, but it did not function properly. There may be other video cards which do work, but none were found.

Many attempts were made to get the system running on some more current hardware, but all attempts failed, for various reasons. A backup of the software was put onto an IDE drive. The software did not pass all tests running on this system (there was a problem with the serial ports used—the original system had 1 serial port, whereas the new one had 2). I could not get the second port disabled. I'm not sure if it would work after doing that or not.

A duplicate hard drive was created on the second AST computer system brought in from Dr. Abel's lab. Although newer, that system doesn't work with the monochrome video card and thus is not suitable as a complete system backup. There are, however, components that can be gleaned from it.

Should the main system fail, this is the recommended procedure to follow:

1) Isolate the problem. Find out which component has failed.
2) If the problem is with the hard drive, the drive from the second AST system can be removed and installed directly into the computer and it should run fine. That hard drive is loaded with the software and should boot up identical to the current system.
3) If the problem is with the motherboard, each of the 4 ISA cards will have to be removed from the current system and moved to the backup AST computer. There was a problem with the video card in this computer, but I believe it may be possible to work around that with some fiddling. Try this out.
4) If the problem is with any of the 4 ISA cards (EGA, GPIB, Serial ports, Colbourne Controller) we have no backups for them.
5) It is possible to try to move everything to the IDE computer. Attempts were relatively successful in doing this, however there were things which did not function properly in this system. Namely, the serial ports seemed to be a problem. There is an ISA card which provides 3 additional serial ports to the computer (for it to communicate with external devices). That card did not function properly in the newer IDE controlled motherboard. Other functions were fine. It may be possible to make that card work. I would start by ensuring that only 1 COMM port was active on the motherboard.

Good luck!!

Appendix E

The MSAccess Noise Database

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33 Introduction

This document provides information pertaining to the Microsoft Access version of the Noise Database. The Access version was based on data extracted from the Macintosh version (i.e. the underlying data was taken from the Macintosh application, however the user interface was re-written in Access).

During the translation from the Macintosh to the PC, several changes to the structure of the data were made. First and foremost, the data was normalized to a greater degree. This allows for a most consistent means of data entry and retrieval. There were other enhancements made and they will be discussed further in this document.

The information contained in this document is an overview of the database. As with any database, the detailed information is contained in the actual database. Please refer to the Access database should you required more detailed information than what is presented in this document. This document should certainly be consulted first, before the database is approached—it will provide invaluable top-level design information required to understand the inner workings of the database.

34 Data Structure

This section deals with the underlying storage containers in which all data is kept. It should be noted that the database contains extra fields which are not always displayed or used. These areas have been created to accommodate future growth.

34.1 AttenuationMeasurements

This table contains the attenuations measured at each frequency, for each particular headset. It contains fields for the mean, standard deviation and number of samples for that measurement.

34.2 Definitions

This table contains the terms and their definitions found within the database.

34.3 FrequenciesOfInterest

This table contains all the frequencies, with their upper and lower boundaries for all the possible measurement types. A measurement type might be 1/3 octave, full octave, etc. There are also columns to allow conversion to/from dBA, dBB and dBC. The final column is called dbLinearTmp and it is used as a temporary storage location for the Tools area of the database. Numbers entered by the user are placed in this location, so that mathematical functions can be performed on the values entered.

34.4 HearingProtectors

This table contains all information pertaining to the hearing protectors in the database. Information relating to the manufacturer and the type of hearing protection are held here.

34.5 HearingProtectorTypes

This table is used to generate select boxes containing a list of all the possible types of hearing protectors (supra-aural, circum-aural, ear plugs, etc...).

34.6 Manufacturers

This table holds the list of all possible manufacturers of hearing protection devices.

34.7 MeasurementFrequencies

This table contains all the information regarding all noise measurements made. As with the AttenuationMeasurements table, the data is normalized, so each record contains information about one frequency of interest made for one measurement.

34.8 MeasurementTeamMembers

This table contains a list of people who are members of a measurement team.

34.9 MeasurementTeams

This table is for future growth and will allow the measurements to be assigned to a measurement team. It is not currently used, however it can be.

34.10 MeaurementTypes

This table contains a list of the possible types of measurements (1/3 octave, full octave, etc...).

34.11 NoiseMeasurements

This table contains the header information relating to all the noise measurements stored in the database.

34.12 Noises

This table contains summary information about all the possible noise sources in the database. Each record in this table can have multiple 'NoiseMeasurements' records, each referencing a measurement perform in this noise.

34.13 NoiseTypes

The NoiseType table holds a short list of the type of noise each record in the 'Noises' table holds. Some possible options are: fixed wing aircraft, rotary wing aircraft, ship, etc...

34.14 People

The People table holds a list of all the people who may be members of a measurement team.

35 User Interface

The interface used to both put data into and retrieve data out of the database is the familiar 'tab' sheets provided by Microsoft Access. This view allows the various areas of the database to be divided into related sections.

Six tabs have been created and they have been purposefully created such that the first three tabs would be used during normal operation of the database, while the last three would be open only to administrators of the database. The intent was that the database could be distributed to anyone, with only the first three tabs accessible. That would allow users to exercise the database, but not make any changes to the underlying data. The last three tabs would only be visible by an administrator, who would make changes to those areas. This idea has not yet been exploited.

35.1 Noise Definitions

The noise definition tab gives the user a quick reference to noise and acoustic related definitions. This screen is simply for viewing of the definitions.

35.2 Noise Tools

The noise tools screen gives the user methods to perform various common tasks with measurement numbers. There are tools to provide the following information:

35.2.1 1/3 Octave to 1/1 Octave Conversion

This tool allows 1/3 octave data to be converted to full octave data.

35.2.2 1/3 Octave Frequencies

This button provides a summary of the frequency ranges associated with a 1/3 octave measurement.

35.2.3 Full Octave Frequencies

This button provides a summary of the frequency ranges associated with a full octave measurement.

35.2.4 DB Conversion

This button provides a tool to allow conversion from any dB'x' to any other dB'y'.

35.2.5 dBA/dBB/dBC weighting factors

This button gives the user a summary page containing the weighting factors associated with dBA, dBB and dBC scaling.

35.2.6 Print All Noise Measurements

This button allows the user to obtain a summary view of all the possible noise measurements contained in the database, relating to one particular noise source.

35.3 Estimate Effectiveness

This tab has a useful tool that allows the user to select a noise source, a location within that noise source and a headset of interest. Given those three parameters, it will predict what the at-ear noise level would be in that situation. The calculations are based on the numbers entered in the database. It provides a good way to perform 'what if' calculations on the noises and headsets.

35.4 Hearing Protectors

This tab allows an administrator to modify data pertaining to headset attenuation measurements. The various parameters of the measurement can be specified, including the levels recorded by the measurement.

35.5 Noise Measurements

This tab allows an administrator to modify data pertaining to a noise measurement made. It can allow for new measurements to be entered as well as old measurements updated.

35.6 Administration

This tab allows for manually entering data into the various tables within the database. It is possible to add new manufacturers, new noise sources, new hearing protectors, etc... through this tab. It is designed to be used by only the administrator.

Appendix F

Calibration of the B&K2133 Direct Inputs

Calibration of the Direct Inputs

On the B&K 2133

This document gives the procedure for calibrating the Direct Inputs on the B&K 2133 Dual Channel Spectrum Analyzer. This procedure should be followed anytime an audio signal is presented to a microphone/pre-amplifier system and its output is delivered to the B&K2133.

When the B&K 2133 is used with a microphone, its sensitivity is known. When it is used with a direct input (microphone and pre-amplifier), its sensitivity is not known. There are 2 specific cases where this procedure must be followed:

- When using the 'ear' mics (for measuring what is under the earcup of a helmet)
- When using the Etymotic microphones with the manikin

First of all, a measurement of a calibrated signal must be made. This is to provide a point to calculate the sensitivity from. The B&K2133 must be set to a sensitivity of 1V/V (unity) when this measurement is made. The following formula can be used to calculate the actual sensitivity that should be used:

$$\text{Sensitivity} = 10^{\left(\frac{MeasuredValue@1Vper1V - CalibrationLevel}{20}\right)}$$

The result of this formula should be entered into the sensitivity field on the B&K2133. To confirm the calculations, the reading obtained with the calibrator in place should read exactly the value from the calibrator.

Step by Step Instructions:

1. Place the microphone in the calibrator
2. Set the sensitivity on the B&K2133 to 1V/V (it is the last parameter on the Channel line of the display (bottom right area of the screen))
3. Take a measurement and note the dB value at the frequency of the calibrator.
4. Put the numbers into the formula and enter the result in the sensitivity field.
5. Run another measurement and ensure the value obtained is that being generated by the calibrator.

Here are a couple of examples:

With the B&K2133 set for 1V/V as the sensitivity, a 124dB calibrator (at 250Hz) generates a 92.8dB signal level at 250Hz. The above formula gives a calculated sensitivity of 27.5mV/V, so 27.5m should be entered. The output then reads 124dB at 250Hz.

With the B&K2133 set for 1V/V as the sensitivity, a 114dB calibrator (at 1000Hz) generates a 73.3dB signal level at 1000Hz. The above formula gives a calculated sensitivity of 9.23mV/V, so 9.23m should be entered. The output then reads 114dB at 1000Hz.

Appendix G

Equipment Manuals

DOCUMENT CONTROL DATA SHEET

1a. PERFORMING AGENCY
Trellis Consulting, 9 Holly Meadow Road, Barrie, Ontario L4N 8S1 (Tel. 905-302-7273)

2. SECURITY CLASSIFICATION
UNCLASSIFIED
-

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Garry Dunn

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Unlimited distribution

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Unlimited announcement

14. ABSTRACT

(U) This document covers various topics relating to acoustic research performed by the Noise and Communication Group at DRDC Toronto. It includes a description of both hardware and software developed and used for research studies conducted in support of Canadian Forces operations. The main components of this document are related to the noise simulation facility and its accompanying software suite, including the Modified Rhyme Test and Maximum Likelihood Procedure. The Microsoft Access Noise Database is also discussed.

(U) Le présent document traite de divers sujets sur les recherches acoustiques menées par le groupe d'acoustique et des communications de RDDC à Toronto. Il comprend une description du matériel et des logiciels conçus et utilisés dans le cadre des recherches techniques effectuées afin d'appuyer les opérations des Forces canadiennes. Les principaux éléments de ce document portent sur l'installation de simulation de bruit et sur la suite logicielle correspondante, notamment la méthodologie de test Modified Rhyme Test et la procédure du maximum de vraisemblance. La base de données Microsoft Access sur le bruit fait également l'objet de discussions.

15. KEYWORDS, DESCRIPTORS or IDENTIFIERS

(U) Noise simulation; speech understanding in noise